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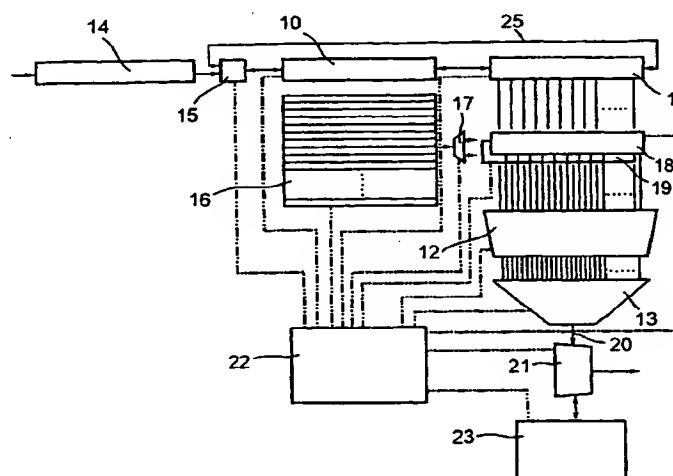
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(54) Title: **METHOD AND DEVICE IN A CONVOLUTION PROCESS**



(57) **Abstract:** A method for convolution of a digital input signal, wherein digital signals corresponding to an impulse response for at least an instance of a spatial environment are stored in a memory (16). Samples of the digital signal are continuously stored in an input buffer (14) and a set of samples, limited in time, of the digital input signal is delimited from continuously incoming samples. The digital signals of the impulse response are divided into a plurality of segments consisting of coefficients and convolution operations are repeatedly performed with a segment of the impulse response and the set of samples, limited in time, of the digital input signal until all segments of the impulse response have been processed. The convolution operations are performed by parallel multiplication of the coefficients of the impulse response and the set of samples, limited in time, of the digital input signal in a multiplier unit (12) and by repeated addition of the products in pairs forming partial results. Partial results are added to previously calculated associated partial results in an adder unit (13) forming an output signal sample.

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METHOD AND DEVICE IN A CONVOLUTION PROCESS

TECHNICAL FIELD OF THE INVENTION

5 The present invention relates to a method in convolution. The method may be used in so called auralising in a space. Auralising implies that a calculation is made about how sounds are interpreted by the listeners both ears take place, so as to recreate the natural experience of space. Stereo sound is part of the sound experience. Instead of two separate microphones, small
10 microphones may be put in the ears of a person being in the recording studio. A person listening to the recorded sound in headphones will experience the same sound image as at the recording occasion. This is called artificial head stereo and is a foundation for auralising, where the headphone sound is created artificially. The invention also relates to a device for performing the
15 method.

 The invention particularly relates to real time auralisation in large premises, for example concert halls. An impulse response for each ear regarding a specific placing and orientation of sound source and listener is calculated in advance. It is also possible to approximately measure the impulse
20 response by firing a pistol where the sound source is going to be and record the sound with microphones in the auditory canals of a so called artificial head on the location of the listener. The time until the impulse response has reverberated is called reverberation time.

 As long as both the sound source and the listener are still the impulse
25 responses are constant. Starting out from these and a non-reverberant recording of the sound source (e.g. a musical instrument on which a certain piece of music is played) the sound for each ear is calculated by an operation called convolution. It forms a filtering of the original sound dependent on both the reflexes (reverberation) in the premises and the direction sensitive
30 influence on sound of the ear before it reaches the eardrum.

 The impulse responses as well as the non-reverberant original sound may be represented as series of numbers and all calculations may take

place digitally, for example in a computer. However, to perform the calculation in real time a great calculation power is required. With a computing rate of 50 kHz (full CD quality) and a reverberation of the room of 2 seconds, $2 \cdot 10^{10}$ (20 billions) multiplications and as many additions per second are required.

STATE OF THE ART

The mathematical definition of convolution for the current application is evident from Formula 1 below.

$$y(t) = \int_0^{\infty} h(v)x(t-v)dv \quad (1)$$

wherein

x constitutes the input,

h constitutes the convolution core (filter) and

y constitutes the output.

Modern signal processing is based on Fourier transforming current signals in the time plane to transformed signals in the frequency plane. The signal processing is then taking place in the frequency plane. The convolution to be performed in the time plane corresponds to multiplication in the frequency plane. As multiplication is a simpler operation, previously signal processing has been implemented by multiplication in fast hardware also in this context.

It is also necessary to transform from the time plane to the frequency plane and back again. For these operations there are algorithms particularly adapted to computers, so called Fast Fourier Transform (FFT), and the discrete equivalence thereof, with inverses. Commercially available processors

adapted to these calculations, so called digital signal processors (DSP), have been present for a long time.

One problem with solutions in the frequency plane is that the calculation time for FFT is growing unpleasantly with the length of the filter ($N \cdot \log N$).

- 5 For convolutions of 100,000 points this method is unfavourable regarding the amount of hardware compared to performing the calculations in the time plane. A contributing cause is that the calculation of FFT requires calculation with floating point numbers, while all calculations in the time plane can be performed with integers. A difficulty with convolution in the time plane is that
- 10 there is no known method to continuously perform such calculations in real time. Neither is there any known device by which the calculations could be performed in real time.

SUMMARY OF THE INVENTION

- 15 One object of the invention is to provide a method for convolution of digital signals. This object is obtained by the invention having the features of claim 1 and 6, respectively. The invention solves the problem of how to perform long convolutions of sound in real time using a reasonably amount of
- 20 hardware.

- According to the invention, the fundamental operations in convolution are performed parallel in an efficient manner. Sound samples from an input signal and terms, or filter coefficients, from the impulse response are stored in registers. Each sound sample and filter coefficient are multiplied with each
- 25 other separately and in parallel. Thereafter the products are added.

- The additions of the products may take place in a so called adder tree, wherein included terms first are added in pairs. The sums again are added in pairs in a repeated sequence until a final sum is calculated. Due to the commutative rule of addition (the order is unessential), this procedure gives exactly the same result as if the original numbers had been added in turn.
- 30

A key question for providing an efficient calculation is how to process the data included in the impulse response. An impulse response may com-

prise in the order of 100,000 samples. To avoid problems with a massive hardware contribution the impulse response is, according to the invention, divided into segments. Required hardware is thereby decreased dramatically because it may be used in a process using time multiplexing.

5 Further, each segment of the impulse response is efficiently used because the convolution operations are performed with the segment together with a plurality of samples from the input signal. The calculated results are added in associated positions in an output buffer, from which an output is fed.

10 Additional advantages and features of the invention are evident from the following description, drawings and dependent claims.

BRIEF DESCRIPTION OF THE DRAWINGS

15 The invention will now be described in more detail with the aid of exemplary embodiments and with reference to the accompanying drawings, in which

Fig. 1 schematically shows an implementation for discrete convolution
20 in the time plane,

Fig. 2 schematically shows one embodiment of an implementation for discrete convolution in the time plane according to the invention,

Fig. 3 shows how registers in the embodiment according to Fig. 2 are co-operating during different phases of the convolution,

25 Fig. 4 shows how contents of registers are changing during different phases of the convolution,

Fig. 5 schematically shows an implementation of an output buffer being used in the embodiment according to Fig. 2, and

30 Fig. 6 shows the function of a calculating unit in the embodiment according to Fig. 2.

DESCRIPTION

A discrete convolution in the time plane may take place according to
5 Formula 2 below. A practical example of implementation is shown in Fig. 1.

$$y(t) = \sum_1^n h(v)x(t-v) \quad (2)$$

wherein

x constitutes input,

10 h constitutes a convolution core (filter) and

y constitutes output.

The basic operations in convolution may be very efficiently paralleled. Input samples are fed into a first shift register 10. A corresponding first filter
15 register 18 holds the impulse response. Sound samples and impulse responses are stored in registers 10 and 18, of which each value has its own direct output. For each pair of sound samples and filter coefficients a separate unit for multiplication is provided, i.e. all multiplications are performed in parallel. Fig. 1 shows all the units for multiplication combined in a multiplier
20 unit 12. All of the results of these multiplications are then to be added, and this may also be performed in a single step in an adder unit 13.

An efficient manner of organising the addition is to add the included numbers in pairs first. Then you get half as many numbers and these may then be added in pairs in a similar manner. As a result you have one single
25 number after a few steps. Thus, a so called adder tree is used. Due to the commutative rule for addition (the order is unessential) you know this is exactly the same result as if the original numbers were added in parallel or in order. As the additions in pairs in each step are performed in parallel the total time is the same as for one, i.e. the total calculation time for the whole

adder tree is proportional to $^2\log N$ instead of N , wherein N is the total number of bits of the included numbers.

Still the delay in the total calculation chain may become too long in proportion to a certain clock cycle time without further measures. This problem may be solved by placing a number of registers on the way and divide the calculation on a number of clock cycles (pipelining).

It is possible to perform the multiplication in a single (combining) step. This will limit the clock frequency and it may be sufficient to place one register before and one after the multiplier, and one inside and one after the adder tree. If an even faster solution is desired the multiplication must also be divided in more pipeline steps.

In the following discussion the starting point is an example having a data transfer rate of 50 kHz for the sound, a clock frequency of 50 MHz for the digital electronics (1000 times faster) and a reverberation time of 2 seconds and a 16 bit data width for both sound and impulse response. An impulse response of 100000 samples is used. By using such high clock frequency it becomes possible to divide the problem into segments and time multiplex the hardware, i.e. make the hardware one thousandth as wide and instead use it 1000 times. Thus, the narrow version of the hardware has exactly the same structure as the original version having the adder tree and pipelining.

A practical embodiment of a convolution unit according to the invention is schematically shown in Fig. 2. The input is introduced through an input buffer 14, suitably arranged as a shift register. The input buffer 14 is connected to a first register 10 through a latch 15. The first register 10 is operatively connected to a second register 11. The first register 10 and the second register are suitably arranged as shift registers. The first register 10 and the second register are fed back through a feedback loop 25 and the latch 15, so that it is possible to shift data back and forth between the registers in a circulating process. The process is described in detail below.

A memory 16 is arranged for storing the impulse response. The impulse response is divided into segments and operates as a filter on the input

signal. In the embodiment shown, the memory 16 is designed for storing 1000 segments, each of 100 samples, or terms. A segment of the impulse response is processed together with a corresponding segment of input. A first calculation step is taking place in a multiplier unit 12. The multiplier unit 12 comprises a plurality of multiplication means for parallel multiplication of specific samples from the input signal and from the impulse response. The segment of the impulse response to be processed is transferred through a multiplexer 17 to a first filter register 18. The memory 16 is also connected to a second filter register 19 through the multiplexer 17.

The embodiment shown in Fig. 2 is particularly suitable for pipelining. Accordingly, included components, e.g. registers and calculation units, comprise a plurality of logical blocks, wherein each block is performing a part of an operation. Several operations in series are thereby performed apparently at the same time.

One filter register is sufficient if the head of the listener is completely still. If the listener is turning the head the same echogram may be used, but new impulse responses must be calculated. This may be performed on a modern PC in a few tens of milliseconds, fast enough to create an apparently continuously active sound image following the head turning in real time.

Also when the head is turning a correct reproduction may be provided by doubling the memory for impulse responses in a similar manner as corresponding buffers in the convolution unit. While one memory is used for convolution the other is filled with new contents. Alternation between the memories may take place momentarily.

Accordingly, while data from the first filter register 18 is processed, new filter data from the impulse response may be transferred to the second filter register 19. Data is alternately used and loaded in the both filter registers 18 and 19, so that the processing may take place without any delay for loading of registers.

In the embodiment according to Fig. 2 and having a suitable clock frequency of the oscillator controlling the electronics, all hardware will be used all the time in continuous operation. One way to make the solution more effi-

cient is to adapt the different calculation units so that an unnecessary number of bits not is used in each case. By this manner it is possible to increase the rate and decrease the amount of hardware.

Each cell of the second shift register 11 and of the filter registers is
5 connected to a specific multiplication means of the multiplier 12, so that the multiplication may take place in parallel. The result from each multiplication is of the length of 32 bits, if the factors included have 16 bits. However, normally only the 11 most significant bits of the result need to be used. It may be even more efficient to adapt the multiplication means, so that they only
10 calculate the bits needed. Then, 11+11 bits are used in a first step of the adder unit 13, which in turn gives a result of 12 bits. Consequently the number of bits is increased with one for each step in the adder tree. Dependent on the number of steps and the number of segments only as many bits as needed are included in the final result.

15 An output 20 of the adder unit 13 is operatively connected to a calculation unit 21. A control unit 22 is operatively connected to the calculation unit 21 and an output buffer 23. The control unit 22 is ensuring that the partial results from the convolution operations available on the output 20 are added to an associated previously calculated partial result stored in the output
20 buffer 23. Suitably the control unit 22 is also controlling remaining components, e.g. the shift registers, the calculation units and the multiplexer. Both the multiplier unit 12 and the adder unit 13 are suitably arranged for pipelining.

The function of the circuit in Fig. 2 may be described schematically in
25 the following way. Sound samples are shifted into the registers, and the different parts of the impulse response are stored in the memory 16, so as to be loaded into one of the filter registers, one part at a time. The register is doubled, one loading while the other is used for convolution and then the operation is alternated (the actual change is taking place between two clock
30 cycles and is not taking any time). Then, as output no longer is produced at the correct rate, an output buffer 23 in the form of a memory having a particular calculation unit is introduced, see the description of Fig. 6 below.

While the filter segment is stored anyway, convolution operations a few points ahead in time (already registered) are performed. This is taking place by the means of a trinomial input buffer comprising the input buffer 14 and the shift registers 10 and 11, in which input data are "swung" back and forth in both the shift registers while new samples are read into the input buffer 14. The results then are superimposed in the correct positions in the output buffer 23 and eventually are fed out in the correct rate.

In connection with the managing of head turning, which is schematically described above, a smaller modification of the controlling of the output buffer is also required, so that the buffer is set to zero only at start up with a new sound and not when new impulse responses are loaded. By this manner a "sliding transition" between filters for two discrete directions is automatically obtained. It is not necessary to change filters more often than what is corresponding to about half of the reverberation time, i.e. once per second in the above described example.

Fig. 3A-3D show how an input may be used in an efficient manner. In a standard position, which is shown in Fig. 3A, new input samples are shifted in through an input 24 of the input buffer 14. The content of the input buffer 14 is then shifted further into the first shift register 10 and the second shift register 11. During the convolution operations the input buffer 14, the first shift register 10 and the second shift register 11 will contain different generations of input data.

During the convolution operations the shift registers are connected to each other in the manner shown in Fig. 3B. The input buffer 14 is separated from the shift registers and is not allowed to change any register content. However, incoming input samples continue to be shifted into the input buffer 14 at the rate they arrive.

While one segment of the impulse response is read into one of the filter registers, e.g. the first filter register 18, input data of the first shift register 10 and the second shift register 11, are "swung" back and forth until all samples are combined with the samples of the impulse response in the first filter register 18. The partial results originated in each position are added to the

associated positions of the output buffer through the calculation unit 21 and are eventually fed out in the correct rate. With a solution adapted to the example all buffers/registers are 100 positions, or samples, long and 16 bits wide and the memory 16 for the impulse response contains 1000 segments.

- 5 Accordingly, the convolution unit is occupied every clock cycle when in a running operation.

In the state shown in Fig. 3C all samples from a segment of the input signal are used. Then the input buffer 14 contains a completely new set, or generation of input data, designated $G(n)$. The first shift register 10 contains the previously used data corresponding to a generation $G(n-1)$ and the second shift register contains even earlier used data, corresponding to a generation $G(n-2)$.

Then, switching into the state shown in Fig. 3D is taking place, in which the content of the input buffer 14 has been transferred to the first shift register 10. This is possible since the latch 15 of the control unit has received instructions to open for the transfer. Then, the first shift register 10 contains the generation $G(n)$ data. The input buffer is set to zero and prepared for introduction of new input samples simultaneously. At the introduction of new data from the input buffer 14 into the first shift register 10 the previously used data is simultaneously shifted from the first shift register 10 to the second shift register 11, which then contains the data of generation $G(n-1)$. Data of the second shift register 11 is not fed into the first register 10, since the feedback loop 25 between the second shift register 11 and the first shift register 10 is broken. This may be accomplished by having the latch 15 breaking the connection.

In the state shown in Fig. 3E the latch 15 has broken the connection between the input buffer 14 and the first shift register 10 again, while the feedback loop 25 is closed once again. A new series of convolution operations may be initiated and a new generation of input $G(n+1)$ is cumulated in the input buffer 14.

Each convolution results in one output point and it has been calculated based on a specific position of the buffer circulating back and forth and a

specific segment of the impulse response. To indicate the position in the data buffer the initial point is that 100 new data just were shifted in, see Fig. 4. This position is called B(0). The next clock impulse gives the position B(1), etc. until B(99). Data is first shifted to the right through the registers until all data from the first register have been shifted into the second register 11. In each position a convolution operation is taking place. After that the same data is used once again by shifting data in the reversed direction. Due to the "swinging" the next position is B(98). Hence, the absolute first convolution is descending from I(0) and B(0). This value is referred to as I(0)B(0). The next value produced is then I(0)B(1) etc. Then, after I(0)B(99) comes I(1)B(98).

When all the segments of the impulse response have been processed, 100 new samples are shifted in and the process starts over from the beginning with I(0)B(0). This value is to be added to the previously stored I(1)B(0). Consequently a designation for the point of time of the input must be introduced to separate the convolution results. The time for the first input buffer is called T(0) etc. With this addition the value for the new output point (no. 100) will be $T(0)I(1)B(0) + T(1)I(0)B(0)$ after updating. Due to the "swinging" the time progress becomes complicated. The comprehension of the calculations may be facilitated by focusing on the result in the output buffer 23.

20

The following designations are introduced:

m the number of elements in the convolution register,

n the number of segments in the impulse response
memory

25

$O(j)$ element in the output buffer

In a so called "steady state", i.e. when the convolution has been in progress for some time, the following Formula 3 applies:

$$O(j) = \sum_{i=0}^{n-1} T(p+i)I(p+i)B(q) \quad (3)$$

wherein

$$p = 0, 1, \dots$$

$$q = 0, 1, \dots, m-1$$

$$j = p + q$$

5

When the convolution is starting all values in the output buffer 23 will be set to zero. As appears in the above table it takes a certain time before all the terms for updating of each output point is available. More specifically will it take exactly as much time as the impulse response is long.

10

In practise the output buffer 23 may be designed as a regular RAM memory, which is organised as a ring buffer in accordance with Fig. 5. The ring buffer comprises an address pointer 26 for start and one address pointer 27 for end. Between start and end the buffer is set to zero. A control unit generates the required addresses so that updating in each moment will take place in the correct position and that the outputting of data takes place in the correct manner. At the same time as a data point is fed out it is set to zero in the ring buffer and becomes thereby prepared for eventually becoming the first value in the buffer again.

15

If the convolution unit produces one result every clock cycle, it will be necessary to read out an old value, add a new value and write back the result to the output buffer during one clock cycle. For example, this may be solved by designing a RAM circuit, so that two values may be reached at a time. Consequently a calculation unit, which is reading, accumulating and writing, must be present between the convolution unit and the memory.

20

Four additions are performed during a process step. As the four additions are not evenly distributed between the four clock cycles, at least one buffer register must be present for intermediate storing of the convolution results from one clock cycle to another. A practical embodiment of such a calculation unit 21 is disclosed in Fig. 6. The calculation unit 21 comprises two parallel pipelines 28 and 29. During four clock cycles the two parallel

25

30

pipelines 28 and 29 are reading and writing, respectively, in two and adding in four.

In clock cycle 1 there is a transfer of values from the output memory to the buffer register #1 and the buffer register #2.

5 In clock cycle 2 a new result value from the convolution unit is added to the value in the buffer register #1. Further, a result value from the convolution unit is added to the value in the buffer register #2. There is also a transfer of values from the output memory to buffer register #3 and buffer register #4.

10 In clock cycle 3 the values in the buffer register #1 and buffer register #2 are transferred to the output buffer 23. A new result value from the convolution unit is simultaneously added to the value in the buffer register #3 and a corresponding addition to the value in the buffer register #4 is made.

Finally, in clock cycle 4 the values in the buffer register #3 and the
15 buffer register #4 are transferred to the output buffer 23. The common control unit 22 is processing these calculations and addresses the memory.

In the above described solution it is assumed that the head of the listener is completely still. If the listener is turning the head the same echogram may be used, but new impulse responses must be calculated. This may be
20 accomplished in a few tens of milliseconds using a modern PC, which is fast enough to create an apparently continuously active sound image following the head turning in real time.

A possible completion to process head turning is to double the memory 16 for impulse responses in a similar manner as the corresponding filter
25 register 18 and 19 in the convolution unit. While one memory is used for convolution the other is loaded with a new content. Alternation between the memories may take place momentarily, so that the convolution does not need to be interrupted. A smaller modification of the control of the output buffer is required so that the buffer is set to zero only at start with a new
30 sound and not when new impulse responses are loaded. By this a "sliding transition" between filters for two discrete directions is obtained. It is not nec-

essary to change filters more often than that corresponding to about half of the reverberation time, i.e. once per second in the described example.

CLAIMS

1. A method for convolution of a digital input signal, *characterised*
5 by
storing digital signals corresponding to an impulse response for
at least an instance of a spatial environment,
continuously storing samples of said digital input signal,
delimiting a set of samples, limited in time, of said digital input
10 signal from continuously incoming samples,
dividing the digital signals of said impulse response into a plural-
ity of segments consisting of coefficients,
repeatedly performing convolution operations to a segment of
the impulse response and the set of samples, limited in time, of the digital
15 input signal until all segments of the impulse response have been processed,
performing said convolution operations by a parallel multiplica-
tion of the coefficients of the impulse response and the set of samples, lim-
ited in time, of the digital input signal and by repeated addition of the prod-
ucts in pairs forming partial results, and
20 adding partial results to previously calculated associated partial
results forming an output signal sample, whereby the convolution is per-
formed in the time plane.
2. A method according to claim 1, wherein the convolution operations are
25 performed by pipelining.
3. A method according to claim 1, wherein the number of coefficients in a
segment of the impulse response is equal to the number of samples in the
set of samples, limited in time, of the digital input signal.
30
4. A method according to claim 1, wherein convolution operations are re-
peatedly performed with a first segment of the impulse response, a second

segment of the impulse response being prepared for the convolution operations simultaneously.

5 5. A method according to claim 1, wherein the partial results from the convolution operations are stored until all coefficients in all segments of the impulse response have undergone convolution operations.

10 6. A device for convolution of a digital input signal, *characterised* in that a memory (16) for storing digital signals corresponding to an impulse response for at least an instance of a space environment, is operatively connected to at least a first filter register (18) that the memory (16) and the first filter register (18) are divided into a plurality of segments, each of which segments having a plurality of cells, wherein each cell may contain a coefficient of the digital signals of the impulse response, that an input buffer (14) is arranged for storing samples of the digital input signal, 20 that the input buffer (14) is operatively connected to memory means (10, 11), that the memory means (10, 11) is designed for gradually shifting data back and forth between cells of the memory means (10, 11), 25 that each cell of the first filter register (18) is connected to a first input terminal of a multiplier unit (12) comprising a plurality of multiplication means, and cells of the memory means (10, 11) are connected to a second input terminal of the multiplication unit (12) for parallel multiplication of the contents of the cells, 30 that an output of each multiplication means is connected to input terminals of an adder unit (13) for addition of the products from the multiplications to a partial result,

that an output of the addition unit (13) is operatively connected to an output buffer (23) through a calculation unit (21),
that the calculation unit (21) comprises addition elements, and
that a control unit (22) is operatively connected to the calculation unit
5 (21) and the output buffer (23), for controlling the calculation unit (21) to add new partial results to previously calculated associated partial results forming an output signal sample.

10 7. A device according to claim 6, wherein the input buffer (14) is operatively connected to a first shift register (10) for transferring a set of samples, limited in time, of the digital input signal, and the first shift register (10) is operatively connected to a second shift register (11) for gradually shifting data back and forth between cells of the first shift register (10) and the second shift register
15 (11).

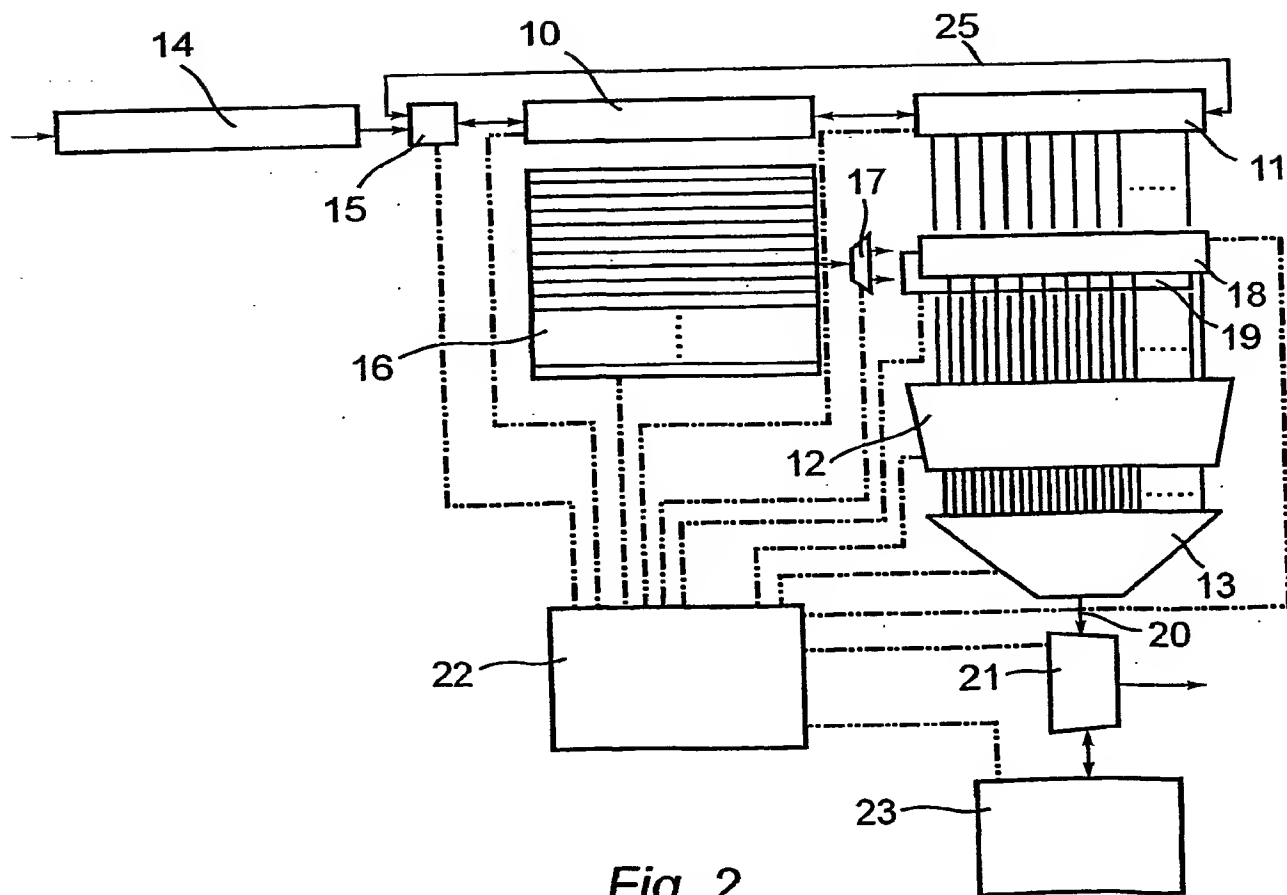
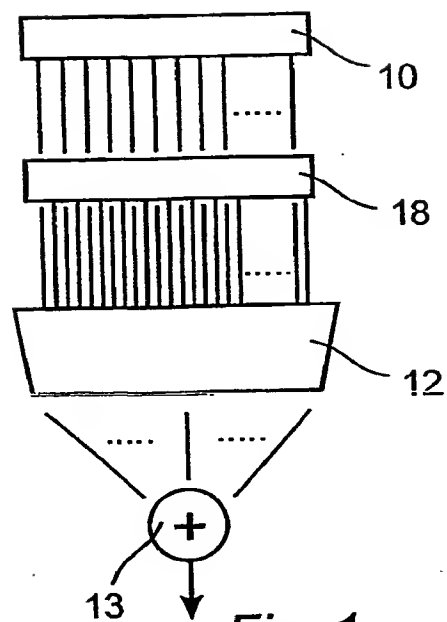
8. A device according to claim 6, wherein the multiplier unit (12) is designed for working with pipelining.

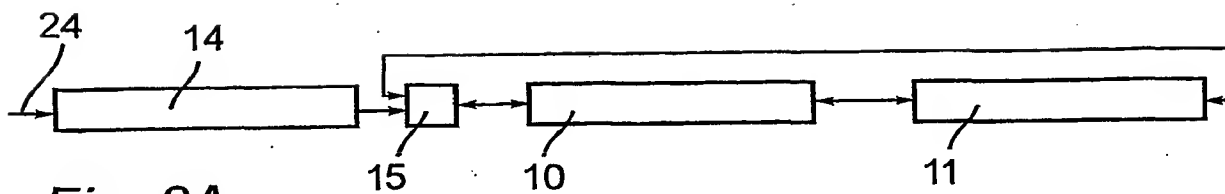
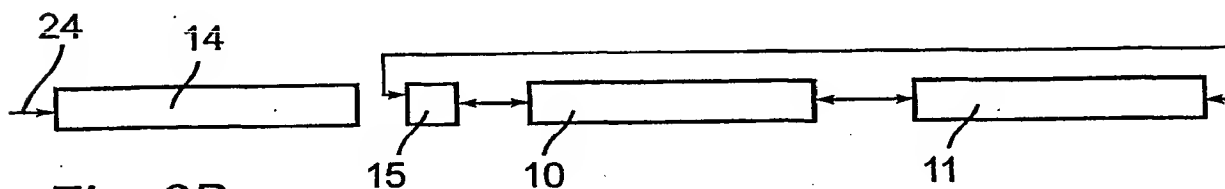
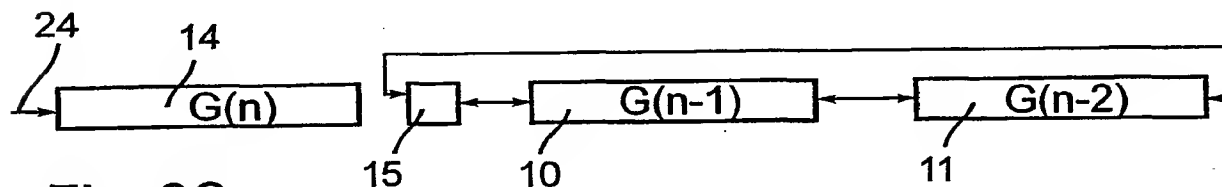
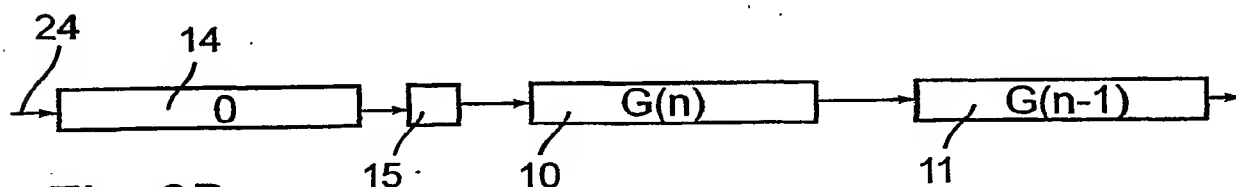
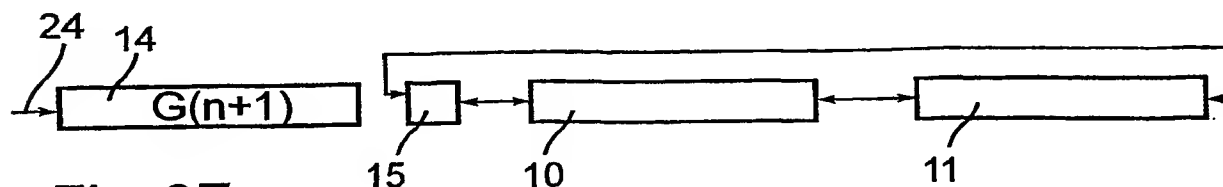
20 9. A device according to claim 6, wherein the adder unit (13) is designed for working with pipelining.

10. A device according to claim 6, wherein the memory (16) is operatively connected to a second filter register (19) for storing coefficients therein, and
25 wherein the first filter register (18) and the second filter register (19) are alternately operatively connected to the multiplier unit (12) and the memory (16), respectively.

11. A device according to claim 6, wherein the memory (16) is doubled to
30 allow convolution without interruptions when changing the impulse response.

1/4



*Fig. 3A**Fig. 3B**Fig. 3C**Fig. 3D**Fig. 3E*

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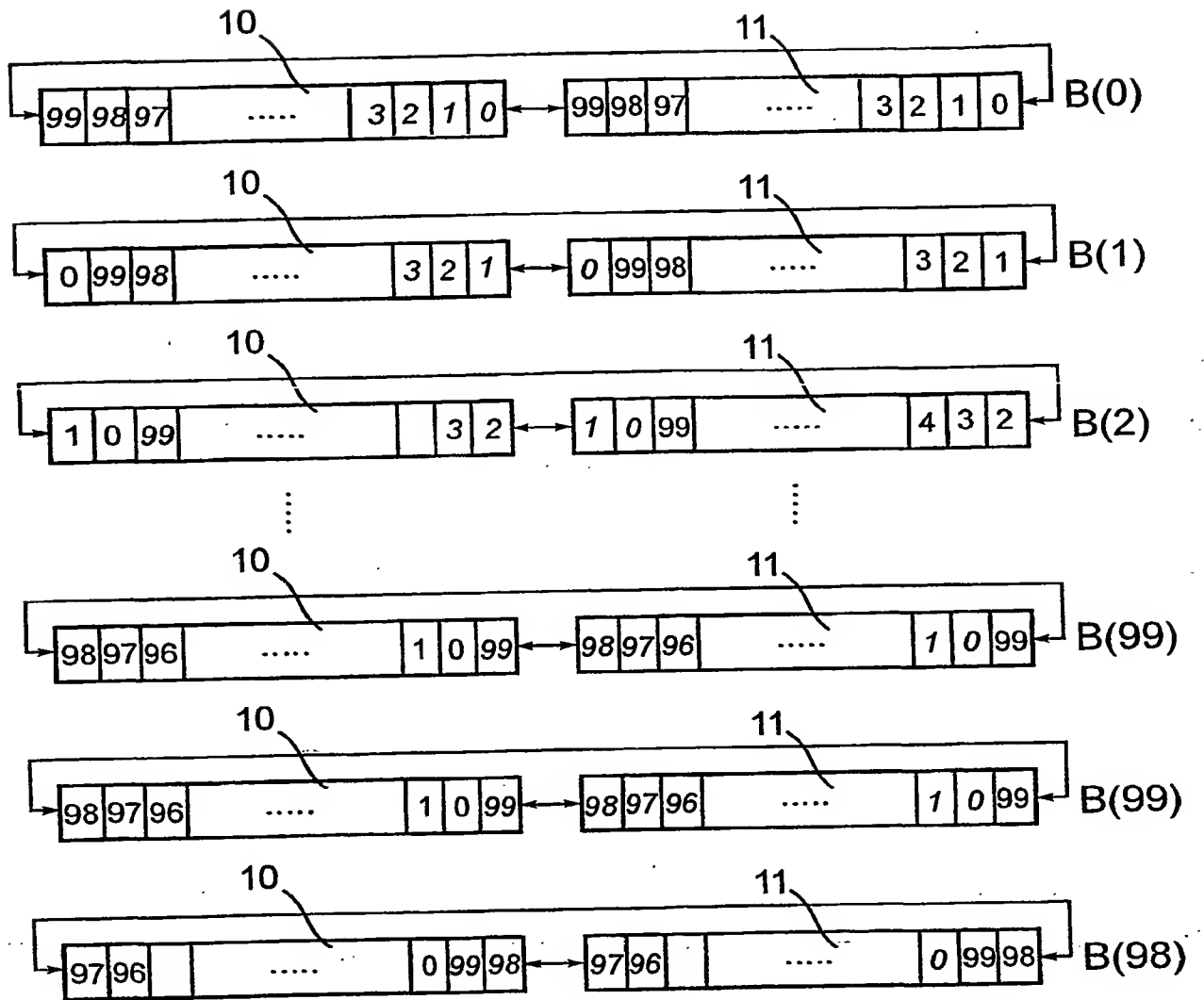


Fig. 4

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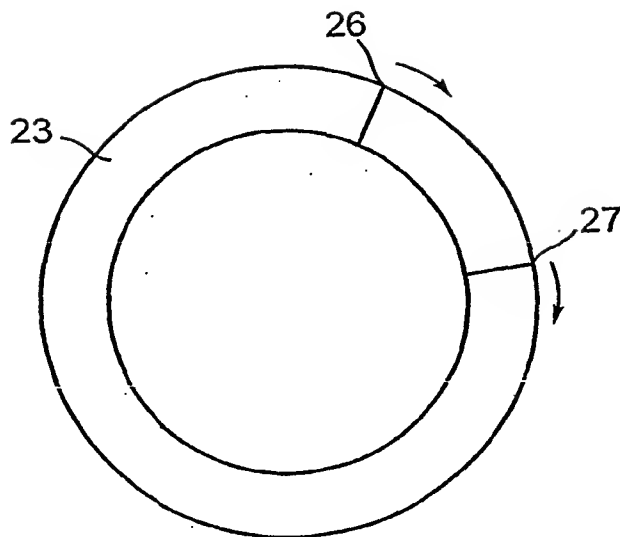


Fig. 5

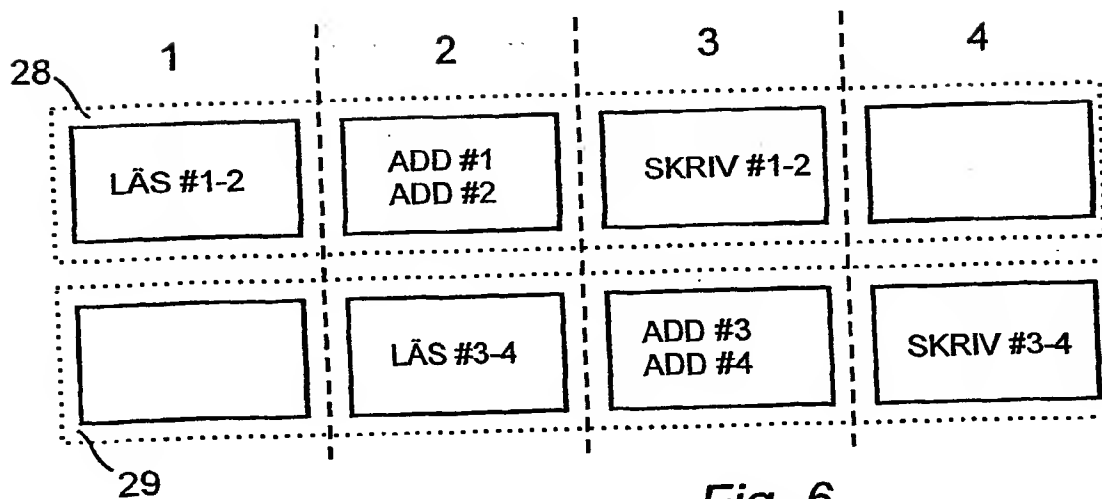


Fig. 6

INTERNATIONAL SEARCH REPORT

International application No.

PCT/SE 01/01074

A. CLASSIFICATION OF SUBJECT MATTER

IPC7: G06F 17/10

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC7: G06F

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

WPI DATA

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 9304529 A1 (KLOKOCA, JIRI), 4 March 1993 (04.03.93) --	1-11
A	US 4862402 A (SHAH, I.A. ET AL.), 29 August 1989 (29.08.89) --	1-11
A	US 5814750 A (WANG, A.L. ET AL.), 29 Sept 1998 (29.09.98) --	1-11
A	US 6000834 A (DUAN, T.), 14 December 1999 (14.12.99) -- -----	1-11

☐ Further documents are listed in the continuation of Box C.☒ See patent family annex.

* Special categories of cited documents:

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"I" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

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"&" document member of the same patent family

Date of the actual completion of the international search

17 Sept 2001

Date of mailing of the international search report

20-09-2001

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INTERNATIONAL SEARCH REPORT
Information on patent family members

03/09/01

International application No.

PCT/SE 01/01074

Patent document cited in search report			Publication date	Patent family member(s)	Publication date
WO	9304529	A1	04/03/93	SE 9102333 D	00/00/00
US	4862402	A	29/08/89	NONE	
US	5814750	A	29/09/98	NONE	
US	6000834	A	14/12/99	NONE	

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